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NATO STANAG 4479: A STANDARD FOR AN 800 BPS VOCODER AND CHANNEL CODING IN HF-ECCM SYSTEM

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ABSTRACT[1]

This paper presents a new voice coder for applications in very low bit rate communication systems normalized by NATO under STANAG agreement 4479. The originality of this standardization is the description of both the source and the channel coding. It is the natural continuation of the well known LPC10e 2400 bps voice coder normalized under STANAG 4198.

The analysis and synthesis are the same as in the LPC10e vocoder but the quantization process is specific. The main points of innovation are presented.

An associated error correcting scheme increases the source bit rate from 800 up to 2400 bps. It has been optimized in the framework of HF-ECCM system (HF-Electronic Counter Counter Measure), to take into account all possible channel perturbations as well as system constraints especially in terms of minimization of the delay and Turn Around Time.

INTRODUCTION

During its March 1994 meeting, NATO group AC302/SG11/WG2 on Narrow Band Speech proposed to ratification the standardization of a 800 bps vocoder for HF communications. The resulting coder and channel protection benefited from a fruitful collaboration with the NSA speech group (T.Tremain, J.Lee). The normalization results from a sophisticated process [1] initiated in 1988, it spans a high quality vocoder associated with a fully optimized error correcting scheme. The NATO standard [2] entails two parts: the firt one is dedicated to the speech coder at 800 bps, the second describes the error correcting code and the interleaver increasing the rate from 800 to 2400 bps in order to protect the system for HF-ECCM. This paper provides a description of the novative points reached in the finalization of the STANAG.

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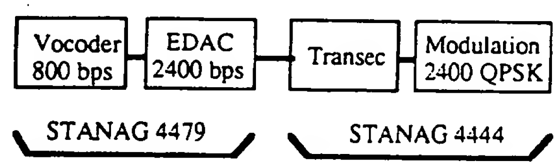
A first part is dedicated to the vocoder and its quantization process. A second part describes the Error Detection And Correction Code (EDAC). The last part deals with other applications of the 800 bps vocoder and its association with other error correcting schemes since very few vocoders [3] have been developped at that rate.

I) THE HF ECCM SYSTEM

The HF-ECCM system is built to allow ciphered and highly protected digitized speech communications on HF links. The carrier modulation (4 Phases Shift Modulation) uses frequency hopping with a slow dwell speed (the carrier frequency varies with time). The system was architectured to be highly resistant to electronic warfare as well as to HF channel perturbations.

Modulation and procedures were already normalized under STANAG 4444. The voice coder and the associated error correcting codes included in the standard 4479 should provide the highest possible quality of service. The interface between the two systems is at 2400 bps.



The 2400 bps voice coder is inadequate for HF frequency hopping systems, since this vocoder with error protecting scheme requires too large a bit rate for this kind of channel (with a serial modulation in frequency hopping it is difficult to transmit more than 2000 to 3000 bps for both vocoder and EDAC). Hence it is necessary to reduce the vocoder bit rate. It resulted from quality and efficiency studies that currently, among many possible rates (1200, 800 bps or 600 bps), the best choice is an 800 bps vocoder. Indeed the quality of the 800 bps is not too far from the 1200 bps but the quality of the 600 bps is juged too low for easy communications. For HF frequency

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hopping perturbations, a redundancy of 2 (with a 1200 bps vocoder) is not sufficient in many cases and a redundancy of 3 is quite better [4], this is why the rate of 800 bps for the vocoder with an additional 1600 bps for the protection has been chosen.

II) THE 800 BPS VOCODER.

For easy transcoding with the LPC10e vocoder, it is much better to keep the same analysis and synthesis in the two coders. The main difference consists in the coding scheme of the parameters resulting from the analysis. An 800 bps coder has not a sufficient bit rate to encode each frame of 22.5ms separetly hence, one will jointly encode a superframe of N successive frames. N=3 is a good compromise between a very efficient rate reduction (large N) and a short vocoder delay (small N). An additional advantage of the choice of a superframe with N=3 is to lead to a vocoder frame of 54 bits, the same size as the LPC10e. This should ease a transparent transmission of the 800 bps through networks conceived for the 2400 bps. The main improvment over previous versions of this coder [5] lies in the Filter encoding and Pitch/ Energy quantization.

II.1 Filter Encoding

The problem is to jointly quantize the filters for the 67.5 ms superframe. Hence the quantization scheme had to be strongly optimized from the 1822 bps (41 bits for each spectrum) used in the LPC10e vocoder. The original solution which requires only 519bps uses eight different coding schemes depending upon the speech stability throughout the superframe. The filter coding is based on the following remarks : if a sound is stable the average spectrum must be well quantized, if it is unstable the emphasis has to be put on spectrum variations rather than on exact values of the filter coefficients. Each scheme corresponds to a particular stability of the signal. The system tries the eight possible schemes and selects the one minimizing a spectral distance (Itakura distance or ceptral distance for instance). It transmits the scheme number on 3 bits and the filter code on 32 bits.

According to the coding scheme, one or two filters will be quantized with 32, 24 or 16 bits. Firstly the vector of the 10 LAR (Log Area Ratio) coefficients which represents the filter is projected on the 10 main axes (eigen vectors of a large data base of spectrum from different languages : English, French, German, Italian, etc...). The projections of the vector on theses 10 axes are then scalar quantized (the rate for each projection is function of the total rate and the rank of the projection). In case one filter is encoded on 24 bits, the 8 additional bits are used to differentially encode the second filter.

II.2 Energy Quantization

10 bits are used to encode each energy in the 3 frames of the superframe. 5 bits are used to encode a reference value (same logarithmic quantization as used in the 2400 bps) which corresponds to the maximum power of the superframe and 5 bits to indicate the power evolution on the superframe (codebook of 32 different evolutions). This is an addition of one bit from previous studies since a more accurate time description of instable sounds is needed when the spectral description is not meaningful.

II.3 Quantization of the Pitch and Voicing configurations

9 bits are used to encode together the pitches and the voicing decisions. Six different voicing configurations can be used for the superframe.

If the superframe is not fully voiced 4 bits indicate the voicing configuration and pitch evolution while 5 bits are used to encode a reference value of the pitches for the whole superframe. If the superframe is fully voiced, 3 bits indicate the voicing configuration and the evolution of the pitch on the superframe (6 different cases) and 6 bits are used to encode the reference value of the pitch.

		2400 bps LPC10	800 bps
Frame length		22.5 ms	22.5 ms
Superframe length		no	67.5 ms
Energy		5 bits	10 bits
Pitch / voicing		7 bits	9 bits
Filter code	Chosen scheme	41 bits	3 bits
	Filter encoding		32 bits
Synchonisation		1 bit	no
Total		54 bits	54 bits

II.4 Speech Quality evaluation

Final DRT tests, organized by the NSA [1] have been conducted by Dynastat. In quiet environment and depending upon the spectral distance taken to decide for the filter quantization scheme, the DRT scores vary around 87. This score is comparable to the LPC10 and not very far from the LPC10e. The listening of the 800 and the LPC10e does not show great differences between the two coders

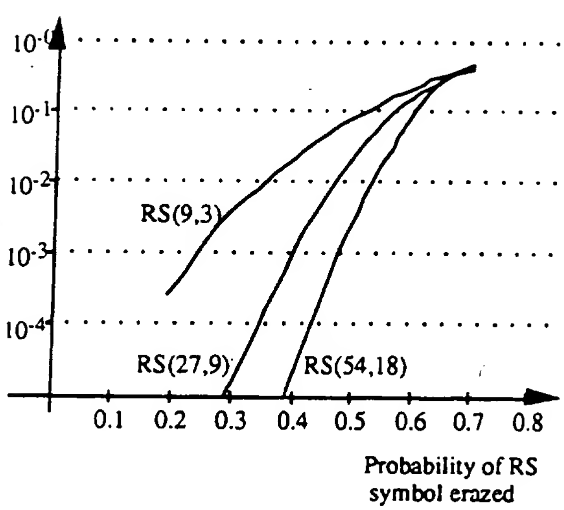
III) ERROR DETECTION AND CORRECTION CODE OF THE 800 BPS VOCODER FOR HF-ECCM.

III.1 Error correcting codes

The selected error correcting code increases the rate from 800 to 2400 bps. Since it is a frequency hopping system, block codes are better than convolutionnal codes to deal with erased dwells without error propagation (in a frequency hopping system some dwells can be completely erazed while some other are unaltered). In addition, since a vocoder frame is much perturbed as soon as one of its protecting block code doesn't decode, it is better to use only one codeword per frame. In such a low bit rate coder all bits have a high sensitivity and the loss of even a small number of bits will have a large influence on the audio quality. Those considerations lead to the final choice of RS(27,9) with 6 bit long symbols, which matches exactly the 54 bits frame.

This is quite a novative approach compared to other vocoders which generally use several block codes. An RS(27,9) code with 6 bit long symbols brings a high degree of protection as shown on the curve below.

Probability of NO RS Decoding

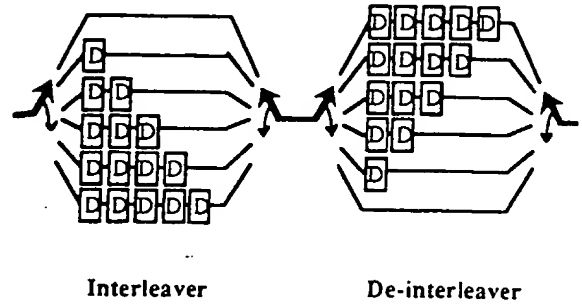


Legend :Average probability of failure in the completly accurate decoding of the vocoder frame given the proportion of erased symbols for various protection schemes. The erasure information is provided by the radio system.

As seen on the above curve, another improvement could have been obtained with a code clustering several coder frames like a RS(54,18) spanning 2 frames. In our application, the longer code the better. The drawback of this solution is to strongly increase the transmission delay (the delay due to the channel coding is doubled).

III.2 Interleaver

An interleaver decreases the influence of block transmission errors but increases the delay. Convolutionnal and Block interleavers were compared. The new use of a convolutionnal interleaver allows a reduction of the transmission delay by a factor two compared to a block interleaver with the same efficiency.



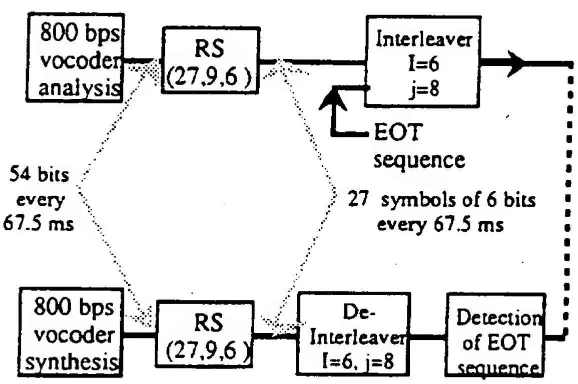
Convolutional interleaver with i= 6 lines and paks
(= D) of j = 8 delais in each line

A well performing convolutionnal interleaver must position 27 symbols of the RS code on a maximum number of dwells and put roughly the same number of symbols on each dwell since the lost of a dwell leads to the erasure of all the symbols it carries (for instance one should avoid to distribute the 27 symbols of a RS code on say 5 dwells with 9,2,2,2 and 12 symbols on each dwell, a more uniform repartition as 5,5,6,6,5 is more robust since no dwell is more sensitive than any other one). A convolutionnal interleaver with a delay of 600 ms with parameter I=6 j=8 was chosen. This 600 ms delay is the highest acceptable to keep the Turn Around Time (TAT) within reasonnable limits. A classical block interleaver would have spanned only half the number of symbols for the same delay which would have sharply decreased the robustness to erased dwells.

III.3 End Of Transmission (EOT)

The EOT detection scheme is an additionnal issue solved by the system. It has to be sufficiently robust to annihilate the Non Detection Probability (NDP), selective enough to avoid False Alarm declaring EOT in the middle of a communication. All this should be done with a minor increase (or better no increase) of the TAT. This problem has been solved by concatenating a sequence of symbols just after the last encoded frame at the entrance of the interleaver.

Due to interleaver delay, an EOT sequence with a duration equal to half the delay (= 120 symbols) is immediatly available when the last vocoder frame has been completly transmitted. On the reception side a correlator positionned upstream from the the de-interleaver allows for the detection of the EOT sequence. Hence the EOT can be detected without any increase of the TAT since the EOT can be detected without de-interleaving. This is a big gain over existing algorithms which lead to an additionnal delay equal to the length of the EOT sequence. Both Symbol and a Bit correlators applied to the output of the convolutionnal encoder have been tested. The symbol correlator has been chosen, actually simulations showed that it is more efficient than the bit correlator. It has a lower false alarm rate for the same non detection rate. Considering that an EOT sequence is detected as soon as 10 symbols (out of 120) match the EOT sequence are identified, the False Alarm Rate is negligible and the NDP low enough for practical use (NDP less than 1.5% even with 40% jammed dwells).



EDAC of the speech mode of HF-ECCM system

IV OTHER APPLICATIONS OF THE 800 BPS VOCODER

The good performances of this coder make it a valuable tool for other communication devices, all the more since it could lead to interoperability between systems.

In fixed frequency HF systems, when associated with an error correcting code and a modem (for instance the STANAG 4285 normalized modem), the 800 bps vocoder allows for a ciphered and secured phone transmission on the HF radio links. The 800 bps vocoder can still be used when the 2400 bps vocoder turns out to be inoperationnal.

The coder can also be used in VHF frequency hopping where the classical bit rate is 16000 bps. Today the CVSD coder Δ16 is currently used. The use of a 800 bps vocoder [6] with a very powerful error correcting code (redundancy up to 20) makes it a "last hope" mode of transmission with a high robustness to electronic warfare and a much increased range compared to the Δ16. In some jamming and bit error conditions the 800 bps transmission can be nearly unaltered while the Δ16 is incomprehensible and deos not even allow to recognize if speech is transmitted or not. The 800 bps vocoder as well as the 2400 bps and the Proprietory ACELP 4800 are implemented in the PR4G, the VHF CNR made by THOMSON.

CONCLUSION

In this paper we described the novel points of the 800 bps vocoder and its surrounding protection scheme. Its quality is close to the LPC10e V52. The re-use of the V52 analysis and synthesis processes makes it simple to implement from the 2400 coder. Hence it can easily be put in lots of communication equipements which is much desired since it is the only communication standard at this rate. It benefits from good system performances resulting from block codes, convolutionnal interleaver and appropriate processing of the EOT which shorten the Turn Around Time.

[1] T. Tremain, D. Kemp, J. Collura, M. Kohler "Evaluation of Low Rater Speech Coders for HF", ICASSP 1993 pp II-163-166.
[2] NATO STANAG 4479 "Parameters and Coding Characteristics that must be common to assure Interoperability of 800 bps Digital Speech Encoder/Decoder and the associated error protection and interleaving schemes leading to the 2400 bps interface", March 1994
[3] D. Kemp, J. Collura, T. Tremain "Multiframe Coding of LPC Parameters at 600-800 bps" ICASSP 1991 pp609-1612
[4] A. Duke, D Rahikka "Error Correction Of Low Rate Speech In Jammed Frequency Hop HF", TCC'92,IEEE/DARPA Tactical Communications Conference April 1992.
[5] B. Mouy, P. de La Noue, "Voice Transmission at a very low bit rate on a noisy channel: 800 bps vocoder with error protection to 1200 bps", ICASSP 1992 pp II-159-152.
[6] A. Vigier, S. Le Dreau "Low rate speech coders for robust VHF radiocommunications", Military Communications, 1993.